IN THE CLAIMS:

This listing of claims will replace all prior versions, and listings, of claims in the application:

LISTING OF CLAIMS:

1. (currently amended) A method for determining whether to accept a new call to be routed from a first location to a second location via a network path in an IP network, comprising the steps of:

(a) obtaining, at the first location, information relevant to the quality of service of voice calls being transmitted from [[a]] the first location to [[a]] the second location via [[an]] the IP network;

(b) calculating, ~~a parameter~~ based on said information, a parameter indicative of a congestion status of the network path from the first location to the second location; and

(c) accepting [[a]] the new call into the IP network at the first location in the case of said parameter not exceeding an upper threshold.

2. (original) The method of claim 1 wherein said new call is accepted into the IP network at a reduced bandwidth in the case of said parameter exceeding a lower threshold.

3. (original) The method of claim 1 where said new call is not accepted into the IP network in the case of said parameter exceeding the upper threshold.

4. (previously presented) The method of claim 1 wherein the information obtained is a number of sent packets transmitted from said first location to said second location in the IP network, wherein the number of sent packets comprises a number of lost packets, a number of late packets and a number of received packets.

5. (original) The method of claim 1 wherein the information obtained is a delay of received packets transmitted from said first location to said second location in the IP network.

6. (original) The method of claim 1 wherein the information obtained is a delay variation of received packets transmitted from said first location to said second location in the IP network.

7. (original) The method of claim 1 wherein the information is obtained on a periodic basis.

8. (original) The method of claim 1 wherein the information is obtained on an exception basis using an immediate report.

9. (original) The method of claim 1 wherein the parameter is identified as a packet lost ratio (PLR).

10. (original) The method of claim 9 wherein PLR is defined as

$$\mathrm{PLR} = \frac{(\text{lost packets} + \text{late packets})}{(\text{received packets} + \text{lost packets} + \text{late packets})}.$$

11. (original) The method of claim 2 wherein bandwidth is reduced for a newly accepted call by selecting a first encoder to encode the new voice call information in a bandwidth that is smaller than bandwidths of other calls accepted in the network that are encoded by a second encoder.

12. (previously presented) The method of claim 2 wherein the bandwidth of a newly accepted call is reduced by increasing the packet size for said newly accepted voice call, wherein the packet size is indicative of a size of a corresponding voice sample.

13. (original) The method of claim 2 wherein the bandwidth of a newly accepted call is reduced by activating the characteristic of silence suppression for said newly accepted voice call.

14. (currently amended) Apparatus comprising a gateway for interfacing voice call data from a public switch telephone network to an internet protocol network,[[;]] said gateway ~~further~~ comprising:

a first circuit for passing said voice call data of voice calls to the internet protocol network;

a second circuit for ~~polling the internet protocol network about traffic information transmitted therein~~ receiving quality-of-service information associated with voice calls currently being transmitted via the first circuit; and

a third circuit for:

calculating, based on the received quality-of-service information, a parameter indicative of a congestion status of a network path associated with the first circuit; and

~~processing the polled information to determine~~ determining, by comparing said parameter to at least one threshold, whether ~~the voice call data~~ a new voice call is to be accepted ~~by~~ into the internet protocol network via the first circuit.

15. (original) The apparatus of claim 14 wherein said first circuit further comprises one or more Ethernet cards that are connected to the internet protocol network.

16. (original) The apparatus of claim 14 wherein said second circuit is at least one strongarm card.

17. (original) The apparatus of claim 16 wherein the strongarm card is connected to the Ethernet card via a host CPU circuit.

18. (currently amended) The apparatus of claim 14 wherein the third circuit ~~compares a parameter based on the polled information~~ determines whether the new voice call is to be accepted into the internet protocol network via the first circuit by comparing said parameter to a plurality of thresholds.

19. (currently amended) The apparatus of claim [[18]] 14 wherein the parameter is a packet loss ratio defined as

$$PLR = \frac{(\text{lost packets} + \text{late packets})}{(\text{received packets} + \text{lost packets} + \text{late packets})}.$$

20. (currently amended) The apparatus of claim 19 wherein the third circuit compares the packet loss ratio to a lower threshold and if the packet loss ratio is less than the lower threshold, [[a]] the new voice call is accepted into the internet protocol network.

21. (currently amended) The apparatus of claim 19 wherein the third circuit compares the packet loss ratio to the lower threshold and an upper threshold, and if lower threshold < packet loss ratio < upper threshold, [[a]] the new voice call is accepted into the internet protocol network at a reduced bandwidth.

22. (currently amended) The apparatus of claim 19 wherein the third circuit compares the packet loss ratio to the upper threshold, and if the packet loss ratio is greater than the upper threshold, [[a]] the new voice call is blocked from entering the internet protocol network.